AT&T IP FlexReach AVPN/MIS/PNT transports: Connecting Cisco Unified Communications Manager Express 8.5 using SIP

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Introduction

Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP FlexReach is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager Express (CISCO UCM E) 8.5 with Cisco Unity Express and Cisco Unified Border Element (CISCO UBE) function for connectivity to AT&T’s IP Flex-Reach SIP trunk service. The application note also covers support and configuration example Cisco Unity Express (CUE) messaging integrated into the Cisco Unified Communications Manager Express. The deployment model covered in this application note is Customer Premises Equipment (CISCO UCME/CUE) to PSTN (AT&T IP Flex-Reach SIP). AT&T IP Flex-Reach provides inbound and outbound call service.

- Testing was performed in accordance to AT&T’s IP Flex-Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, DNIS translations, CODEC negotiation, advanced SYY call prompter, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Express), Cisco auto-attendant (BACD), fax using T.38 and G.711 (G3 and SG3 speeds), teleconferencing, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.

- The Cisco Unified Border Element function configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Unified Communications Express. The configuration described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying CISCO UBE to ensure these commands are set per each dial-peer required, to interoperate to AT&T SIP network.

- For IP Flexible Reach service with AVPN access, the CUCME LAN IP address (facing the AT&T CER) can be an AT&T assigned signaling IP address. For IP Flexible Reach service with MRS/PNT (and optionally with AVPN) access, the CUCME LAN IP address (facing the CER) can be private IP address. This will be NAted by the AT&T managed CER (or customer managed/MRS managed CER for AVPN). Consult with AT&T provisioning engineer to resolve any IP addressing issues.

- Please refer to the Emergency 911/E911 Limitations and Restrictions section of this document for more information on Emergency 911/E911 services.

- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications Express with Cisco Unified Border Element components.
The Cisco Unified Communication Manager Express depicted in Figure 1 also functions as a border element. It is not an AT&T managed device.
System Components

**Hardware Components**

- Cisco Integrated Service Router G2. This solution was tested with C3925 but this application note applies to any ISR G2 platform. Refer to the following link for more information on ISR G2 platforms:
  

- Cisco IP Phones. This solution was tested with 7961 and 7975 phones, but any Cisco IP Phone model supporting RFC2833 can be used.

- Cisco SRE (Service Ready Engine) compatible with ISR G2 platform to run Cisco Unity Express application. This solution was tested with ISM-SRE-300-K9 but the following modules may be used:
  
  - Internal Service Module (ISM) – ISM-SRE-300-K9
  - Service Module (SM) – SM-SRE-700-K9 or SM-SRE-900-K9
  - Enhanced Network Module (NME) - NME-CUE (will need SM-NM-ADPTR adapter)

  Please refer to the following CUE Compatibility Matrix link for information on hardware, platforms, minimum IOS software and service/network module requirements.


- Packet Voice Data Module (PVDM). You will need to install DSP modules (PVDM) on the ISR G2 platform if you require MTP, Transcoding or Conference Bridge resources for codecs other than G.711. DSPs are not required for basic calls. Follow the following link for system required DSP calculator.

  [http://www.cisco.com/cgi-bin/Support/DSP/cisco_dsp_calc.pl](http://www.cisco.com/cgi-bin/Support/DSP/cisco_dsp_calc.pl)

**Software Requirements**

- Cisco IOS gateway running Cisco Unified Communication Manager Express (CUCME) 8.5 (IOS version 15.1.3T release). This solution was tested with Cisco IOS image: C3900-universalk9-mz.SPA.151.3.T.bin

  - Cisco Unified Border Element (CUBE) is an integrated Cisco IOS software application that requires a separate CUBE feature license to connect to AT&T SIP trunk.

  - Cisco Unity Express is an integrated Cisco IOS software application that requires a separate UNITY feature license to implement voicemail and other messaging features. This solution was tested with Cisco Unity Express version (8.0)

Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications components. For reference, please follow this link:

Features

Features Supported

- Basic Call using G.729
- Calling Party Number Presentation and Restriction
- Calling Name
- AT&T Advanced 8YY Call Prompter (8YY)
- Intra-site Call Transfer
- Intra-site Conference (See Caveat section for details)
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax using T.38 (See Caveat section for details)
- Fax over G.711 (See Caveat section for details)
- Incoming DNIS Translation and Routing
- CISCO UBE: performs Delayed-Offert-to-Early-Offert conversion of an initial SIP INVITE without SDP
- Outbound calls to AT&T’s IP and TDM networks
- CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (Cisco Unity Express)
- Auto-attendant transfer-to service (See Caveat section for details)
- Failover (From non-responsive SIP network to legacy PSTN circuit)

Features Not Supported

- CISCO UCME/CISCO UBE Codec negotiation of G.722.1
- Real-Time Transport Control Protocol (RTCP) (See Caveat section for details)
**Caveats**

Fax:
- The maximum fax rate achieved using T.38 (G3 or SG3) fax protocol is only 9600 kbps. This is related to Cisco defect id CSCtk06444, where max bit rate is not updated and fails to advertise V.17 capability.
- An issue was found when receiving T.38 Fax at SG3 speed over certain local PSTN NPAs served by the legacy SBC platform. This is related to Cisco defect id: CSCoo45871 where the CUCME fails to send a T.38 re-invite.

Auto-Attendant:
- The CUCME Basic Automatic Call Distribution (BACD) was employed to enable the auto-attendant feature. The test was performed using the default codec G711 for auto attendant prompts. G.729 prompts can be used; however it was not tested here.

Real-Time Transport Control Protocol (RTCP)
- CUCME does not support the periodic transmission of RTCP sender report to provide statistics of RTP flow. Certain AT&T networks such as AVPN require this feature.

**Configuration considerations**

- When using G.729 between AT&T IP Flex-Reach and Cisco Unified Communication Manager Express (CUCME)/Cisco Unified Border Element (CUBE) SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between G729 media end-points. See configuration section for details.
- For forwarded calls from CUCME user to PSTN (out to AT&T’s IP Flex-reach service) some AT&T serviced areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption was made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using CISCO UCM translation patterns. Because we use 4-digit extensions on our CUCME IP phones, it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message, to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in CUBE (See configuration section for details.).
- Upon receiving inbound calls, AT&T SIP network will always select the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place/receive G.711-only calls must configure separate dial-peer(s) on CUCME/CUBE with G.711 codec assigned. Typically, this solution is used for fax transmissions using G711.
- Some SIP components within AT&T core do not support the “:0” as the Boolean value within the “T38FaxFillBitRemoval” parameter within the SDP header of a fax Re-INVITE. Thus, a sip profile is used to remove this attribute to achieve fax T.38 interoperability across AT&T SIP core.
- SIP Profiles may also be employed to advertise desired RTP payload packet size.
Emergency 911/E911 Services Limitations and Restrictions

- Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

Configuration

Cisco IOS version

CME-G2-SP#sh ver
Cisco IOS Software, C3900 Software (C3900-UNIVERSALK9-M), Version 15.1(3)T, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2010 by Cisco Systems, Inc.
Compiled Tue 16-Nov-10 01:48 by prod_rel_team

ROM: System Bootstrap, Version 15.0(1r)M6, RELEASE SOFTWARE (fc1)

CME-G2-SP uptime is 1 minute
System returned to ROM by power-on
System image file is "flash0:c3900-universalk9-mz.SPA.151-3.T.bin"
Last reload type: Normal Reload

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:

If you require further assistance please contact us by sending email to export@cisco.com.

Cisco CISCO3925-CHASSIS (revision 1.0) with C3900-SPE100/K9 with 989184K/59392K bytes of memory.
Processor board ID FTX1436A0X0
4 Gigabit Ethernet interfaces
1 terminal line
1 Virtual Private Network (VPN) Module
4 Voice FXS interfaces
1 Internal Services Module (ISM) with Services Ready Engine (SRE)
DRAM configuration is 72 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
500472K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:
License UDI:

<table>
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<th>Device#</th>
<th>PID</th>
<th>SN</th>
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<tbody>
<tr>
<td>*0</td>
<td>C3900-SPE100/K9</td>
<td>FOC14341EZJ</td>
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Technology Package License Information for Module: 'c3900'

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<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
</tr>
<tr>
<td>data</td>
<td>datak9</td>
<td>Permanent</td>
<td>datak9</td>
</tr>
</tbody>
</table>

Configuration register is 0x2102
Cisco Unified Communication Manager Express (CUCME)

CME-G2-SP#sh run
Building configuration...

Current configuration : 16745 bytes
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CME-G2-SP
!
boot-start-marker
boot system flash0:c3900-universalk9-mz.SPA.151-3.T.bin
boot-end-marker
!
!
card type t1 1 1
logging buffered 999999
logging rate-limit console 10000
no logging console
enable password cisco
!
no aaa new-model
!
!
crypto pki token default removal timeout 0
!
!
no ipv6 cef
ip source-route
ip cef
!
!
!
!
no ip domain lookup
!
multilink bundle-name authenticated
!
!
isdn switch-type primary-ni
!
!
voice-card 0
dspfarm
This command enables DSP farming allowing DSP resources for Media Termination Point (MTP), Conference Bridge (CFB) or Transcoder.

2 Enables IP address hiding between the private network (CUCME side) and the public network (AT&T FlexReach side)

3 This command enables the CUCME with CUBE feature to perform basic IP to IP voice communication

4 This command enables T.38 fax at global level, meaning all VoIP dial-peers not configured for specific fax protocol will use this setting. T.38 fax protocol may be configured under appropriate dial-peers.

5 This command enables the use of Early Offer SIP INVITE method

6 This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and Cisco Unified Communications Manager Express (CUCME)

7 This command allows the CUCME to negotiate all flavors of G729 codec and must be configured in order to interoperate seamlessly across AT&T’s BVoIP services. The command can either be enabled globally, as in this example, or per dial-peer basis using the “voice-class sip g729 annexb-all” command

8 This command enables multiple codec support and performs codec filtering required for correct interoperability between AT&T SIP network and Cisco UCME. Payload packet size can also be configured here.

9 SIP Profiles can be used to manipulate SIP header attributes

10 This SIP profile expands the Diversion header number from a 4-digit extension to a full 10-digit DID number in order to attain interoperability with AT&T’s HIPS (LEGACY) served users for forwarded calls.

11 This SIP Profile allows CUCME to advertise desired and supported payload packet size
request INVITE sdp-header Audio-Attribute add "a=ptime:30"
request REINVITE sdp-header Attribute modify "a=T38FaxFillBitRemoval:0" 

voice class custom-cptone CONF
dualtone conference
frequency 600 900
cadence 300 150 300 100 300 50
!
!

voice register global
mode cme
source-address 172.20.110.157 port 5060
max-dn 100
max-pool 185
dialplan-pattern 1 .... extension-length 4
tftp-path flash:
create profile sync 0069050054835118
!
!!

voice translation-rule 1
rule 1 /^.*(....)/ /7322161/
!

voice translation-rule 2
rule 2 /^.*(....)/ /1/
!

voice translation-rule 3
rule 3 /3143323714/ /2714/

voice translation-profile NPA
    translate calling 1
!

voice translation-profile 10DigitTo4
    translate called 2
!

voice translation-profile LEGACY
    translate called 3
!

application
service aa flash:app-b-acd-aa-2.1.2.3.tcl
    paramspace english index 0
    param menu-timeout 3
    param handoff-string aa
    param dial-by-extension-option 3

---

12 This SIP profile removes the SDP attribute “T38FaxFillBitRemoval:0” from Cisco IOS gateway upspeed Re-INVITE (inbound call to CPE.) Some SIP components within AT&T’s SIP core do not support the “:0” as the Boolean value, instead some AT&T devices interpret the full attribute as the Boolean value (1=attribute present; 0=attribute not present). For this reason, we remove the attribute completely to achieve fax t.38 interoperability across AT&T’s entire SIP core.

13 Enables the CUCME BACD auto attendant feature.
paramspace english language en
param operator 6001
param max-time-vm-retry 2
param aa-pilot 6999
param max-extension-length 4
paramspace english location flash:
param second-greeting-time 60
param welcome-prompt _bacd_welcome.au
param welcome_prompt_en_bacd_welcome.au
param call-retry-timer 15
param voice-mail 5000
paramspace english prefix en
param max-time-call-retry 60
param service-name queue
!
!
!
license udi pid C3900-SPE100/K9 sn FOC14341EZJ
hw-module ism 0
!
hw-module pvdm 0/0
!
!
username cisco privilege 15 secret 5 $1$Cj/.SdU5cRW4lb02gUU.zgc1ME0
!
redundancy
!
!
controller t1 1/0
pri-group timeslots 1-24
!
translation-rule 1
!
!
interface Loopback0
ip address 172.1.1.1 255.255.255.0
!
interface GigabitEthernet0/0
description SETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
speed auto

interface ISM0/0
  ip unnumbered GigabitEthernet0/0
  service-module ip address 172.20.110.158 255.255.255.0
!Application: CUE Running on ISM
  service-module ip default-gateway 172.20.110.1
!
interface GigabitEthernet0/1
  description connection to ATT Network
  ip address 70.132.127.108 255.255.255.0
  duplex auto
  speed auto
!
interface GigabitEthernet0/2
  no ip address
  duplex auto
  speed auto
!
interface Serial1/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-ni
  isdn incoming-voice voice
  no cdp enable
!
interface ISM0/1
  description Internal switch interface connected to Internal Service Module
  shutdown
!
interface Vlan1
  no ip address
!
!
  ip forward-protocol nd
!
  no ip http server
  no ip http secure-server
!
  ip route 172.20.0.0 255.255.0.0 172.20.110.1
  ip route 172.20.110.158 255.255.255.0 ISM0/0
  ip route 207.242.225.200 255.255.255.255 70.132.127.110
  ip route 207.242.225.210 255.255.255.255 70.132.127.110
!
logging esm config
access-list 101 permit ip host 207.242.225.200 host 70.132.127.108
access-list 101 permit ip host 207.242.225.210 host 70.132.127.108
access-list 101 permit ip host 70.132.127.108 host 207.242.225.200

14 Cisco Unity Express Interface
access-list 101 permit ip host 70.132.127.108 host 207.242.225.210
!  
!  
!  
!  
nls resp-timeout 1  
cpd cr-id 1  
  
!  
tftp-server flash0:dsp75.9-1-1TH1-16.sbn  
tftp-server flash0:jar75sccp.9-1-1TH1-16.sbn  
tftp-server flash0:SCCP75.9-1-1SR1S.loads  
tftp-server flash0:term75.default.loads  
tftp-server flash0:apps75.9-1-1TH1-16.sbn  
tftp-server flash0:cnu75.9-1-1TH1-16.sbn  
tftp-server flash0:cvm75sccp.9-1-1TH1-16.sbn  
tftp-server flash0:jar41sccp.9-1-1TH1-16.sbn  
tftp-server flash0:SCCP41.9-1-1SR1S.loads  
tftp-server flash0:term41.default.loads  
tftp-server flash0:term61.default.loads  
tftp-server flash0:apps41.9-1-1TH1-16.sbn  
tftp-server flash0:cnu41.9-1-1TH1-16.sbn  
tftp-server flash0:cvm41sccp.9-1-1TH1-16.sbn  
tftp-server flash0:dsp41.9-1-1TH1-16.sbn  
!  
control-plane  
!  
!  
voice-port 0/1/0  
!  
voice-port 0/1/1  
!  
voice-port 0/1/2  
!  
voice-port 0/1/3  
!  
voice-port 1:0:23  
!  
!  
scp local GigabitEthernet0/0
!  
scp ccm 172.20.110.157 identifier 1 version 7.0  
scp  
!  
scp ccm group 100  
  
bind interface GigabitEthernet0/0  
associate ccm 1 priority 1  
associate profile 2 register con1cdf0fafe00  
associate profile 1 register MTP1cdf0fafe00  
!  

These sccp commands configure the shared DSP resources as conference bridge (CFB) and transcoder device for the CUCME.
dspfarm profile 1 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  codec g729r8
  codec g729br8
  codec g722-64
  maximum sessions 3
  associate application SCCP

! dspfarm profile 2 conference
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  codec g729r8
  codec g729br8
  codec g722-64
  maximum sessions 4
  conference-join custom-cptone CONF
  associate application SCCP

! dial-peer voice 1999 voip
description OUTGOING CALL TO AT&T FACING AT&T NETWORK
translation-profile outgoing NPA
destination-pattern 1T
session protocol sipv2
  session target ipv4:207.242.225.210
  incoming called-number 732........
  voice-class codec 4
  voice-class sip asserted-id pai
  voice-class sip privacy-policy passthru
  voice-class sip early-offer forced
  voice-class sip profiles 2
  dtmf-relay rtp-nte
  fax rate 14400
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  no vad

! 16 This command sets the SIP server target for outgoing SIP calls.
  17 This command assigns the voice class codec setting to this dial-peer.
  18 This command enables the delivery of caller id information using P-asserted-ID method, across the SIP trunk. This command can either be issued globally or per dial peer.
  19 This command allows for privacy settings to be transparently passed between AT&T network and CUCME. This command can either be issued globally or per dial peer.
  20 This commands enables Early Offer SIP invite method.
  21 This commands assigns the applicable SIP profile to use for this dial-peer.
  22 Example of configuring T38 as fax protocol per dial peer.
dial-peer voice 19991 voip
description ALTERNATE ROUTE FOR OUTGOING CALL TO AT&T FACING AT&T NETWQRK
preference 1
translation-profile incoming NPA
translation-profile outgoing NPA
destination-pattern 1T
session protocol sipv2
session target ipv4:1.1.1.1
incoming called-number 732....... voice-class codec 4
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 19992 voip
description ALTERNATE ROUTE TO PSTN
preference 2
destination-pattern 1T
direct-inward-dial
translate-outgoing calling 1
port 1/0:23
forward-digits all
!
dial-peer voice 1998 voip
description OUTGOING CALL TO BENCH PHONE VIA AT&T LEGACY NETWORK
destination-pattern 3143463905
translation-profile outgoing NPA
session protocol sipv2
session target ipv4:207.242.225.210
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
fax rate 14400
no vad
!
dial-peer voice 1997 voip
description OUTGOING CALL TO BENCH PHONE VIA AT&T CISCO GW
destination-pattern 9142223902
translation-profile outgoing NPA
session protocol sipv2
session target ipv4:207.242.225.210
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
dtmf-relay rtp-nte
fax rate 14400
no vad
!
dial-peer voice 1141 voip
description OUTGOING INTERNATIONAL CALL TO AT&T FACING AT&T NETWORK
translation-profile outgoing NPA
destination-pattern 0114158330158
session protocol sipv2
session target ipv4:207.242.225.210
incoming called-number 314........
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
dtmf-relay rtp-nte
fax rate 14400
no vad
!
dial-peer voice 511 voip
description OUTGOING N11 CALL TO AT&T FACING AT&T NETWORK
translation-profile outgoing NPA
destination-pattern 911
session protocol sipv2
session target ipv4:207.242.225.210
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
fax rate 14400
no vad
!
dial-peer voice 1000 voip
description INCOMING DIAL PEER FACING AT&T NETWORK
translation-profile incoming 10DigitTo4
session protocol sipv2
incoming called-number 732216....
voice-class codec 4
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
fax rate 14400
no vad
!
dial-peer voice 1996 voip
description INCOMING DIAL PEER FROM AT&T LEGACY NETWORK
translation-profile incoming LEGACY
destination-pattern 1T
session protocol sipv2
session target ipv4:207.242.225.210
incoming called-number 3143323714
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
codec g722-64
fax rate 14400
!
dial-peer voice 4715 pots
description ONLY USED FOR FAX TEST CALLS
shutdown
destination-pattern 2715
port 0/1/2
!
dial-peer voice 4717 pots
description ONLY USED FOR FAX TEST CALLS
shutdown
destination-pattern 2717
port 0/1/3
!
dial-peer voice 8715 voip
description OUTGOING AND INCOMING FAX OVER G711 CALL TO/FROM AT&T FACING AT&T NETWORK
translation-profile outgoing NPA
destination-pattern 1T
session protocol sipv2
session target ipv4:207.242.225.210
incoming called-number 7322162715
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
playout-delay nominal 80
playout-delay mode fixed
codec g711ulaw
no fax-relay sg3-to-g3
fax rate 14400
no vad
!
dial-peer voice 8717 voip
description INCOMING FAX OVER G711 CALL FROM AT&T LEGACY NETWORK
translation-profile incoming LEGACY
destination-pattern 1T
session protocol sipv2
session target ipv4:207.242.225.210
incoming called-number 3143323714
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
playout-delay nominal 80
playout-delay mode fixed
codec g711ulaw
no fax-relay sg3-to-g3
no vad
!
!
sip-ua
mwi-server ipv4:172.20.110.158 expires 3600 port 5060 transport tcp unsolicited
!
!
!
!
gatekeeper
shutdown
!
!
telephony-service
sdspfarm units 5
sdspfarm transcode sessions 128
sdspfarm tag 1 con1cdf0f0afe00
sdspfarm tag 2 MTP1cdf0f0afe00
conference hardware
no auto-reg-ephone
em logout 0:0 0:0 0:0
max-ephones 96
max-dn 192
ip source-address 172.20.110.157 port 2000
service phone g722CodecSupport 2
load 7961 sccp41.9-1-1SR1S.loads
load 7962 SCCP42.8-5-4S.loads
load 7975 SCCP75.9-1-1SR1S.loads
voicemail 5000
mwi relay
max-conferences 4 gain -6
call-forward pattern .T
moh music-on-hold.au
web admin system name cisco password cisco
dn-webedit
time-webedit
transfer-system full-consult
transfer-pattern .T
secondary-dialtone 9
create cnf-files version-stamp Jan 01 2002 00:00:00
!
ephone-template 1
  softkeys idle Cfw dall ConfList Dnd Gpickup HLog Join Login Newcall Pickup Redial RmLstC
  softkeys seized Redial Pickup Gpickup HLog Meetme Endcall
  softkeys connected Acct ConfList Confrn Endcall Flash HLog Hold Join Park RmLstC Select Trnsfer
!
!
ephone-dn 1 dual-line
  number 2714
  label Phone A
  name Phone A
  call-forward busy 5000
  call-forward noan 5000 timeout 10
  translate calling 1
!

ephone-dn 2 dual-line
  number 2715
  label Phone B
  name Phone B
!

ephone-dn 3 dual-line
  number 2716
  label Phone C
  name Phone C
!

ephone-dn 4 dual-line
  number 2717
  label Phone D
  name Phone D
!

ephone-dn 5 dual-line
  number 3143323714
!

ephone-dn 6 dual-line
  number 3143323715
  label Phone F
  name Phone F
!

ephone-dn 10 dual-line
  number 6245
  name Chadi Testing
!

ephone-dn 30 dual-line
number 3333
conference meetme
preference 1
no huntstop
!
!
ephone-dn 31 dual-line
number 3333
conference meetme
preference 2
no huntstop
!
!
ephone-dn 32 dual-line
number 3333
conference meetme
preference 3
no huntstop
!
!
ephone-dn 33 dual-line
number 3333
conference meetme
preference 4
!
!
ephone-dn 34 dual-line
number 4444
name conference
conference ad-hoc
no huntstop
!
!
ephone-dn 35 dual-line
number 4444
name conference
conference ad-hoc
preference 1
no huntstop
!
!
ephone-dn 36 dual-line
number 4444
name conference
conference ad-hoc
preference 2
no huntstop
!
!
ephone-dn 37 dual-line
number 4444
name conference
classification ad-hoc
preference 3
!
!
ephone-dn 80
number 8000....
mwi on
!
!
ephone-dn 81
number 8001....
mwi off
!
!
ephone 1
device-security-mode none
mac-address 001D.705F.B383
ephone-template 1
codec g722-64
type 7975
button 1:1 2:5
!
!
ephone 2
device-security-mode none
mac-address 001B.0CDB.B399
ephone-template 1
codec g722-64
type 7961
button 1:2 2:6
!
!
ephone 3
device-security-mode none
mac-address 0017.0EEE.3279
ephone-template 1
codec g722-64
type 7961
button 1:3
!
!
ephone 4
device-security-mode none
mac-address 001B.0CDB.CB27
ephone-template 1
codec g722-64
type 7961
button 1:4

!  
!  
ephone 5
device-security-mode none
codec g722-64
!  
!  
ephone 6
device-security-mode none
codec g722-64
!  
!  
ephone 10
device-security-mode none
mac-address 0023.0433.CF07
codec g722-64
type 7962
button 1:10
!  
!
sip-ua
retry invite 2
no remote-party id
!  
!
line con 0
password cisco
login
stopbits 1
line aux 0
stopbits 1
line 195
no activation-character
no exec
transport preferred none
transport input all
transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
line vty 0 4
access-class 23 in
exec-timeout 15 0
privilege level 15

23 This command should be enabled when failover dial-peers are configured.
password cisco
login
transport input telnet
!
exception data-corruption buffer truncate
scheduler allocate 20000 1000
end

Cisco Unity Express (CUE)

se-172-20-110-158# sh run

Generating configuration:

clock timezone UTC
hostname se-172-20-110-158
line console

system language preferred "en_US"

ntp server 172.20.110.157 prefer

software download server url "ftp://127.0.0.1/ftp" credentials hidden
"6u/dKTNhEeUfBw40XIf2eFHnZfyUT5d8Z2Ngd+Y9J3xIk2B35j0nGWTYHfmPd8Z2Ngd+Y9J3xIk2B35j0nGWTYHfmP"

license agent max-sessions 9

privilege ViewRealTimeReports create
privilege ViewPrivateList create
privilege manage-passwords create
privilege manage-users create
privilege ViewHistoricalReports create
privilege ManagePublicList create
privilege vm-imap create
privilege local-broadcast create
privilege ManagePrompts create
privilege broadcast create

groupname Broadcasters create

username PhoneA create
username cisco create
username Bruno create
privilege ViewRealTimeReports description "Privilege to view realtime reports"
privilege ViewPrivateList description "Privilege to view private list"
privilege manage-passwords description "Privilege to reset user passwords"
privilege manage-users description "Privilege to create, modify, and delete users and groups" privilege ViewHistoricalReports description "Privilege to view historical reports"
privilege ManagePublicList description "Privilege to manage public lists"
privilege vm-imap description "Privilege to manage personal voicemail via IMAP client"
privilege local-broadcast description "Privilege to send local broadcast messages"
privilege ManagePrompts description "Privilege to create, modify, or delete system prompts"
privilege broadcast description "Privilege to send local or remote broadcast messages"
privilege ViewRealTimeReports operation report.realtime
privilege ViewPrivateList operation voicemail.lists.private.view
privilege manage-passwords operation user.pin
privilege manage-passwords operation user.password
privilege manage-passwords operation system.debug
privilege manage-users operation user.mailbox
privilege manage-users operation user.remote
privilege manage-users operation user.pin
privilege manage-users operation user.password
privilege manage-users operation user.notification
privilege manage-users operation group.configuration
privilege manage-users operation system.debug
privilege manage-users operation user.configuration
privilege ViewHistoricalReports operation report.historical.view
privilege ManagePublicList operation voicemail.lists.public
privilege ManagePublicList operation system.debug
privilege vm-imap operation voicemail.imap.user
privilege local-broadcast operation broadcast.local
privilege local-broadcast operation system.debug
privilege ManagePrompts operation prompt.modify
privilege ManagePrompts operation system.debug
privilege broadcast operation broadcast.remote
privilege broadcast operation broadcast.local
privilege broadcast operation system.debug

groupname Administrators member cisco
groupname Broadcasters privilege broadcast

username PhoneA phonenumber "2714"

restriction msg-notification create
restriction msg-notification min-digits 1
restriction msg-notification max-digits 30
restriction msg-notification dial-string preference 1 pattern * allowed

backup server url "ftp://127.0.0.1/ftp" credentials hidden
"EWlTygcMhYmjazXhE/VNXHCKpLVV4KjescbDaLa4fl4WLSPFvv1rWUnfGWTYHfmPsd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPsd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"
calendar biz-schedule system
schedule
open day 1 from 00:00 to 24:00
open day 2 from 00:00 to 24:00
open day 3 from 00:00 to 24:00
open day 4 from 00:00 to 24:00
open day 5 from 00:00 to 24:00
open day 6 from 00:00 to 24:00
open day 7 from 00:00 to 24:00
end schedule

ccn application autoattendant aa
description "autoattendant"
enabled
maxsessions 10
script "aa.aef"
parameter "dialByExtnAnytime" "false"
parameter "busOpenPrompt" "AABusinessOpen.wav"
parameter "dialByExtnAnytimeInputLength" "4"
parameter "operExtn" "1001"
parameter "welcomePrompt" "AAWelcome.wav"
parameter "disconnectAfterMenu" "false"
parameter "dialByFirstName" "false"
parameter "busClosedPrompt" "AABusinessClosed.wav"
parameter "allowExternalTransfers" "false"
parameter "holidayPrompt" "AAHolidayPrompt.wav"
parameter "businessSchedule" "systemschedule"
parameter "MaxRetry" "3"
end application

ccn application ciscomwiapplication aa
description "ciscomwiapplication"
enabled
maxsessions 10
script "setmwi.aef"
parameter "CallControlGroupID" "0"
parameter "strMWI_OFF_DN" "8001"
parameter "strMWI_ON_DN" "8000"
end application

ccn application msgnotification aa
description "msgnotification"
enabled
maxsessions 10
script "msgnotify.aef"
parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
parameter "DelayBeforeSendDTMF" "1"
end application

ccn application promptmgmt aa
description "promptmgmt"
enabled
maxsessions 10
script "promptmgmt.aef"
parameter "appManagementScript" ""
end application

ccn application voicemail aa
description "voicemail"
enabled
maxsessions 10
script "voicebrowser.aef"
parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
end application

ccn engine
end engine

ccn reporting historical
database local
description "se-172-20-110-158"
end reporting

ccn subsystem jtapi
ccm-manager address 0.0.0.0
end subsystem

ccn subsystem sip
gateway address "172.20.110.157"
end subsystem

ccn trigger http urlname msgnotifytrg
application "msgnotification"
enabled
maxsessions 2
end trigger

ccn trigger http urlname mwiapp
application "ciscomwiapplication"
enabled
maxsessions 1
end trigger

ccn trigger sip phonenumbe 5000
application "voicemail"
enabled
maxsessions 2
end trigger

service voiceview
enable
end voiceview

voicemail broadcast recording time 300
voicemail default messagesize 240
voicemail notification restriction msg-notification
voicemail mailbox owner "PhoneA" size 300
expiration time 10
messagesize 120
end mailbox

end
se-172-20-110-158#
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>CISCO UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CISCO UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>SP</td>
<td>Service Provider</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public switched telephone network</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>TDM</td>
<td>Time-division multiplexing</td>
</tr>
<tr>
<td>CODEC</td>
<td>Coder-Decoder (in this document a device used to digitize and undigitize voice signals)</td>
</tr>
</tbody>
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